Amendments to the Specification

Please replace the title of the invention with the following new title:

INSERTING AUXILIARY DATA IN AN AUDIO DATA STREAM

Please amend the paragraph beginning at page 7, line 24 as follows:

The filterbanks in MPEG audio have the property of (nearly) perfect reconstruction. A diagram of a decoder to an encoder is shown in Fig. 1. If the filterbanks (102, 104) are aligned correctly then the subband samples (106) in the encoder will be practically identical to those (108) that originated in the decoder.

Please amend the paragraphs beginning at page 8 (Amended Sheet), line 10 as follows:

Fig. 2 shows the measured level (202) of the audio in each subband, coded as "scalefactors" in the MPEG audio bitstream. It also shows the bit allocation (204) chosen by an encoder. This is specified as the number of quantisation levels for a particular subband. In the diagram, the bit allocation is represented as a signal-to-noise ratio, in dB terms, to permit representation on the same axis. For this purpose, each bit that is needed to represent the number of quantisation levels is approximately equivalent to 6dB of "level".

If instead we show the scalefactors (302) and the lowest level that can be encoded (304) with the bit allocation from Fig. 2 we get the graph in Fig. 3.

Please amend the paragraph beginning at page 9, line 7 as follows:

If we are decoding an MPEG bitstream to insert data, we would not know the level of that subband so, to be safe, we should probably not send any data in that subband. If, on the other hand, we are using an encoder purely for generating data we could use the levels just below the full level in this subband. A diagram showing the area where the data could be inserted (402), for the latter case, is shown in Fig. 4.

Please amend the paragraphs beginning at page 17, line 7 as follows:

An example synchronisation sequence (602), shown in Fig. 6, consists of a sine wave with certain points set to zero. This can be inserted into an upper subband, e.g. subband 30. For 48kHz sampling this is above the maximum subband (27) defined by the MPEG standard. Thus this extra synchronisation signal would not be coded by a "dumb" encoder.

This sequence (700 of Fig. 7) should be inserted into an appropriate subband before the synthesis filter ~~(see Fig. 7)~~ (702 of Fig. 7). The analysis filter would then produce subband samples from which the frame and 32-sample boundary can be deduced.